IN THE CLAIMS

Please amend the claims to read as follows:

1. (Currently Amended) A method for conducting call fallback in a gateway, comprising:

receiving an incoming call;

establishing a Voice over IP (VoIP) call over a VoIP network;

generating audio packets from the incoming call and sending the audio packets over the VoIP call;

monitoring quality of service on the VoIP network during the VoIP call;

setting up a fallback call over a circuit switched network during the VoIP call when the monitored quality of service of the VoIP network degrades;

cross connecting the incoming call to both the fallback call and the VoIP call during midcall of the VoIP call after the fallback call has been setup and sending same audio data over both fallback call and the VoIP call at the same time; and

redirecting the incoming call from the currently established VoIP call to the fallback call.

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2. (Original) A method according to claim 1 including:

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terminating the VoIP call;

continuing monitoring quality of service on the VoIP network during the fallback call;

establishing a new VoIP call during the fallback call when the quality of service of the VoIP network improves;

cross connecting the incoming call to both the fallback call and the new VoIP call when the new VoIP call is established;

sending audio packets encoded from the incoming call over the new VoIP call; and

terminating the fallback call.

3. (Original) A method according to claim 2 including signaling that the new VoIP call is associated with the fallback call by identifying the fallback call in headers of the audio packets sent over the new VoIP call.

4. (Original) A method according to claim 1 including:

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establishing the fallback call over either an ISDN fallback call having an outgoing ISDN channel or a DS0 fallback call having an outgoing PSTN DS0 channel;

receiving the incoming call over an incoming PSTN DS0 channel;

cross connecting the incoming call for a DS0 fallback call by cross connecting the incoming PSTN DS0 channel to the outgoing PSTN DS0 channel and then terminating the generation of audio packets from the incoming call;

cross connecting the incoming call for an ISDN fallback call by continuing generation of audio packets from the incoming call and rerouting the audio packets over the outgoing ISDN channel.

5. (Original) A method according to claim 1 wherein setting up the fallback call comprises:

detecting when the quality of service for the VoIP call has degraded below a predetermined level;

identifying any already established ISDN call with a same destination gateway as the VoIP call;

determining if the identified ISDN call has bandwidth capacity for carrying the VoIP call; and

using the ISDN call identified with sufficient bandwidth capacity as the fallback call for the VoIP call.

6. (Original) A method according to claim 1 including:

generating audio packets for multiple incoming calls;

setting up the fallback call as an ISDN connection;

cross connecting the audio packets for the multiple incoming calls over that same ISDN connection at the same time.

7. (Original) A method according to claim 1 including:

receiving in an input buffer the audio packets from the VoIP call;

receiving audio signals from the fallback call and at the same time receiving the audio packets from the VoIP call; and

disconnecting the VoIP call when the audio packets in the input buffer have been played out.

8. (Original) A method according to claim 1 including:

receiving a new incoming call;

identifying any previously received incoming calls that are currently cross connected over fallback calls to a destination gateway targeted for the new incoming call; and

preventing the new incoming call from being established as a new VoIP call when any previously received incoming calls are identified.

9. (Original) A method according to claim 1 wherein setting up the fallback call includes sending signaling in the fallback call that associates the fallback call with the VoIP call.

10. (Original) A method according to claim 9 wherein the signaling comprises ISDN signaling, SS7 signaling or DTMF signaling.

11. (Original) A method according to claim 1 including:

sending time stamps in the audio packets of the VoIP call;

identifying a fallback call receive time when the fallback call is received at a destination gateway;

comparing the packet time stamps with the identified fallback call receive time; and

cross connecting the fallback call to an output of the destination gateway when one of the time stamps converge with the fallback call receive time.

12. (Currently Amended) A VoIP gateway, comprising:

a telephony interface for receiving an incoming call;

a VoIP interface for encoding the incoming call into audio packets;

and sending the audio packets over a VoIP network; and

a cross connect that sets up a fallback call over the telephone interface and cross connects the incoming call to the fallback call sending audio data over the fallback call at the same time that the same audio signals for the incoming call ~~is~~ are being encoded into audio packets and sent over the VoIP call.

13. (Original) A VoIP gateway according to claim 12 wherein the cross connect receives a quality of service measurement for the VoIP network and cross connects the incoming call to the fallback call when the quality of service of the VoIP network degrades.

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14. (Original) A VoIP gateway according to claim 12 wherein the cross connect:

receives a termination signal back from a destination gateway to terminate the VoIP call after audio signals from the incoming call has been received by the destination gateway over the fallback call;

terminates the VoIP call by disabling the VoIP interface from encoding and sending the audio packets from the incoming call over the VoIP network;

monitors the quality of service on the VoIP network during the fallback call after the VoIP call has been terminated;

establishes a new VoIP call when the quality of service improves;

reenables the VoIP interface to encode and send audio packets from the incoming call over the new VoIP call; and

terminates the fallback call after receiving confirmation from the destination gateway that the audio packets are being received.

15. (Original) A VoIP gateway according to claim 12 wherein the fallback call is established as an ISDN fallback call having an outgoing ISDN channel or an outgoing PSTN DS0 fallback call having an outgoing PSTN DS0 channel.

16. (Original) A VoIP gateway according to claim 15 wherein:

the incoming call is received over an incoming PSTN DS0 channel;

the cross connect cross connects the incoming call directly to the outgoing PSTN DS0 channel when the fallback call is the PSTN DS0 fallback call; and

the VoIP interface reroutes the audio packets encoded for the incoming call to the outgoing ISDN channel when the fallback call is the ISDN fallback call.

17. (Original) A VoIP gateway according to claim 12 wherein the cross connect establishes the fallback call over the telephony interface as follows:

detecting when the quality of service for the VoIP call has degraded below a predetermined level;

identifying any existing ISDN fallback call on the telephony interface currently established with a destination gateway that is also receiving the VoIP call;

determining whether the existing ISDN fallback call has bandwidth available for carrying the VoIP call;

using any identified ISDN fallback call with available bandwidth as the fallback call for the VoIP call; and

establishing a new fallback call on the telephony interface when there is no identified ISDN fallback call already established with the destination gateway having available bandwidth capacity.

18. (Original) A VoIP gateway according to claim 15 wherein the telephony interface receives multiple different incoming calls all directed to the same destination gateway and the cross connect cross connects the audio packets for the multiple different incoming calls over the same outgoing IDSN channel.

19. (Original) A VoIP gateway according to claim 12 wherein the cross connect prevents any new incoming call from being sent over the VoIP network when any fallback call is directed to a same destination gateway as the new incoming call.

20. (Currently Amended) A VoIP gateway, comprising:

a VoIP interface receiving audio ~~packet~~ <u>packets</u> from an incoming VoIP call, decoding the audio packets, and outputting the decoded audio packets to an outgoing telephony call;

a telephony interface for receiving a fallback call during the in-progress VoIP call <u>containing same audio data contained in the audio packets</u> and receiving signaling that associates the fallback call with the VoIP call; and

a cross connect that cross connects the fallback call to the outgoing telephony call of the associated VoIP call and drops the VoIP call after receiving audio signals over the fallback call.

21. (Original) A VoIP gateway according to claim 20 including a buffer for receiving the audio packets, the cross connect dropping the VoIP call after the VoIP interface has decoded all the audio packets in the buffer.

22. (Original) A VoIP gateway according to claim 20 wherein an operating system in the VoIP gateway monitors the quality of service of the VoIP network and sends a signal to

an originating gateway to initiate the fallback call when the quality of service of the VoIP network degrades.

23. (Original) A VoIP gateway according to claim 20 wherein the cross connect:

sends a termination signal back to an originating gateway to terminate the VoIP call after receiving a voice signal from the originating gateway through the fallback call;

continues monitoring the quality of service on the VoIP network during the fallback call after the VoIP call has been terminated;

sends a request to the originating gateway to establish a new VoIP call when the quality of service of the VoIP network improves; and

terminates the fallback call after the new VoIP call has been established and VoIP packets from the new VoIP call have been received.

24. (Original) A VoIP gateway according to claim 20 wherein the fallback call is either an ISDN fallback call or a PSTN DS0 fallback call.

25. (Original) A VoIP gateway according to claim 20 wherein;

the fallback call contains audio packets;

the VoIP interface decodes the audio packets from the fallback call; and

the cross connect cross connects the decoded audio packets from the fallback call to the telephony interface.

26. (Original) A VoIP gateway according to claim 20 wherein:

the telephony interface receives VoIP packets from multiple different incoming calls on the same fallback call;

the VoIP interface decodes the VoIP packets from the different incoming calls on the same fallback call; and

the cross connect cross connects the decoded audio packets from that same fallback call to different outputs on the telephony interface.

27. (Currently Amended) A computer readable medium for use with a network processing device, the computer readable medium comprising:

detecting an incoming call;

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initiating a Voice over IP (VoIP) call over a VoIP network for the incoming call that generates audio packets from the incoming call and sends the audio packets over the VoIP call;

monitoring quality of service on the VoIP network during the VoIP call;

initiating a fallback call over a circuit switched network during the VoIP call when the monitored quality of service of the VoIP network degrades;

cross connecting the incoming call to both the fallback call and the VoIP call during midcall of the VoIP call after the fallback call has been setup and sending same audio data over both the fallback call and the VoIP call at the same time; and

redirecting the incoming call from the currently established VoIP call to the fallback call.

28. (Previously Presented) A computer readable medium according to claim 27 including:

initiating termination of the VoIP call;

maintaining monitoring quality of service on the VoIP network during the fallback call;

initiating establishment of a new VoIP call during the fallback call when the quality of service of the VoIP network improves;

cross connecting the incoming call to both the fallback call and the new VoIP call when the new VoIP call is established, audio packets encoded from the incoming call sent over the new VoIP call; and

initiating termination of the fallback call.

29. (Previously Presented) A computer readable medium according to claim 28 including identifying the new VoIP call with the fallback call using headers in the audio packets sent over the new VoIP call.

30. (Previously Presented) A computer readable medium according to claim 27 including:

initiating establishment of the fallback call over either an ISDN fallback call having an outgoing ISDN channel or a DS0 fallback call having an outgoing PSTN DS0 channel;

detecting receipt of the incoming call over an incoming PSTN DS0 channel;

cross connecting the incoming call for a DS0 fallback call by cross

connecting the incoming PSTN DS0 channel to the outgoing PSTN DS0 channel and then terminating the generation of audio packets from the incoming call; and

cross connecting the incoming call for an ISDN fallback call by continuing generation of audio packets from the incoming call and rerouting the audio packets over the outgoing ISDN channel.

31. (Previously Presented) A computer readable medium according to claim 27 wherein setting up the fallback call comprises:

detecting when the quality of service for the VoIP call has degraded below a predetermined level;

identifying any already established ISDN call with a same destination gateway as the VoIP call;

determining if the identified ISDN call has bandwidth capacity for carrying the VoIP call; and

using the ISDN call identified with sufficient bandwidth capacity as the fallback call for the VoIP call.

32. (Previously Presented) A computer readable medium according to claim 27 including:

generating audio packets for multiple incoming calls;

setting up the fallback call as an ISDN connection;

cross connecting the audio packets for the multiple incoming calls over that same ISDN connection at the same time.

33. (New) A system for use with a network processing device, the computer readable medium comprising:

means for detecting an incoming call;

means for initiating a Voice over IP (VoIP) call over a VoIP network for the incoming call that generates audio packets from the incoming call and sends the audio packets over the VoIP call;

means for monitoring quality of service on the VoIP network during the VoIP call;

means for initiating a fallback call over a circuit switched network during the VoIP call when the monitored quality of service of the VoIP network degrades;

means for cross connecting the incoming call to both the fallback call and the VoIP call during midcall of the VoIP call after the fallback call has been setup and sending same audio data over both the fallback call and the VoIP call at the same time; and

means for redirecting the incoming call from the currently established VoIP call to the fallback call.

34. (New) The system according to claim 33 including:

means for initiating termination of the VoIP call;

means for maintaining monitoring quality of service on the VoIP network during the fallback call;

means for initiating establishment of a new VoIP call during the fallback call when the quality of service of the VoIP network improves;

means for cross connecting the incoming call to both the fallback call and the new VoIP call when the new VoIP call is established, audio packets encoded from the incoming call sent over the new VoIP call; and

means for initiating termination of the fallback call.

35. (New) The system according to claim 34 including identifying the new VoIP call with the fallback call using headers in the audio packets sent over the new VoIP call.

36. (New) The system according to claim 33 including:

means for initiating establishment of the fallback call over either an ISDN fallback call having an outgoing ISDN channel or a DS0 fallback call having an outgoing PSTN DS0 channel;

means for detecting receipt of the incoming call over an incoming PSTN DS0 channel;

means for cross connecting the incoming call for a DS0 fallback call by cross connecting the incoming PSTN DS0 channel to the outgoing PSTN DS0 channel and then terminating the generation of audio packets from the incoming call; and

means for cross connecting the incoming call for an ISDN fallback call by continuing generation of audio packets from the incoming call and rerouting the audio packets over the outgoing ISDN channel.

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37. (New) The system according to claim 33 wherein the means for setting up the fallback call comprises:

means for detecting when the quality of service for the VoIP call has degraded below a predetermined level;

means for identifying any already established ISDN call with a same destination gateway as the VoIP call;

means for determining if the identified ISDN call has bandwidth capacity for carrying the VoIP call; and

means for using the ISDN call identified with sufficient bandwidth capacity as the fallback call for the VoIP call.

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